

## METHOD FOR ACOUSTIC TRANSDUCER CALIBRATION

## TECHNICAL FIELD

5           This invention relates in general to acoustic calibration and more specifically acoustic calibration for speaker and microphone anomalies as used in communications equipment.

## BACKGROUND

10           Many portable communications devices use some variety of transducer. A transducer can include such devices as a microphone to convert acoustic energy to electrical energy or a speaker to convert the electrical energy back to acoustic energy. Ideally, it is important to achieve some type of predetermined frequency response and gain from these devices in order for the communications device to operate most effectively. A transducer with a wide frequency response enables a complete spectrum of audio frequencies to be reproduced which are typically between 300 to 3000 Hertz

20           (Hz). However, the acoustic responses of these transducer devices unfortunately are non-ideal, inconsistent and often have poor operational characteristics. This is due to such things as environmental factors, the mechanical placement of the transducer and/or variations in their manufacture.

25           For example, a typical microphone used in a two-way radio device often can have a gain of +/- 3 decibel (dB) as specified by most manufacturers. In the design and operation of two-way radio or cellular devices, this can make it difficult to electrically balance audio to the input

30           circuitry of the device. This is due to wide variations in both microphone gain and frequency response. This same example is also applicable to the communications speaker output which often causes a user using numbers of similar types of communications equipment difficulty in maintaining

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a similar operating radio when comparing two devices. More often than not, this causes the user to falsely determine that a radio is defective when in-fact only slight acoustic variations in operation between either microphone or speaker  
5 cause each radio to sound differently to the user.

Therefore, the need exists to provide a system for acoustic microphone and speaker calibration that will enable an electronic device to operate consistently regardless of slight operational dissimilarities between the microphone  
10 and speaker components.

#### BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram showing acoustic calibration  
15 of a microphone in a portable communications device.

FIG. 2 is a block diagram showing the method of acoustic calibration of a microphone according to the preferred embodiment of the invention.

FIG. 3 is a block diagram showing the acoustic  
20 calibration of an internal speaker in a portable communications device.

FIG. 4 is a block diagram showing the method of acoustic calibration of an internal speaker according to the preferred embodiment of the invention.  
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#### DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring now to FIG. 1, a portable two-way communications device 101 such as a two-way radio or  
30 cellular telephone includes an internal speaker and internal microphone 103. In the preferred embodiment of the invention, during the acoustic calibration of a microphone 103, a characterized external speaker 105 is attached to the communications device 101 that is used to produce audible

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pseudo random noise generated by an internal digital signal processor (DSP). The pseudo random noise is directed toward the microphone 103. As is well known in the art, acoustic band limited pseudo random noise is often referred to as  
5 "pink noise" and is audio generated over the audible frequency range of 300 Hz to 3KHz.

FIG. 2 depicts a block diagram showing the method of acoustic calibration of the microphone 103 according to the preferred embodiment of the invention. Pseudo random noise  
10 201 is generated and supplied to a filter 203. The pseudo random noise can be generated either internally from the communications device or from an external source. The filter 203 acts to tailor the frequency response of the external speaker 105 in order to provide optimized frequency  
15 and gain characteristics for microphone calibration where "h" is the frequency response of the speaker and "1/h speaker" is the inverse frequency response. 1/h speaker is used to denote the combination of frequency responses to produce a "flat" frequency response. Thus, filter 203  
20 effectively normalizes the frequency and gain response of the speaker 105 used for calibration of the microphone 103. DSP 209, as discussed hereinafter, is the actual device the optimizes the characteristics of microphone 103.

The amplitude of the pseudo random noise coming from  
25 speaker 105 is sufficient enough such that it is supplied to the input of microphone 103. Although microphone 103 is shown as an internal microphone, it will be evident to those skilled in the art the an external speaker microphone, such as a speaker microphone, could be calibrated using this  
30 method as well. The output of the microphone 103 is directed to a digital signal processor (DSP) type audio filter 209. As is well known in the art, the DSP 209 acts to transform the analog microphone input and convert it to a digital signal where it can be easily processed and

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manipulated to add, remove or alter its signal characteristics. These signal characteristics include but are not limited to amplitude or frequency components.

In order to control the DSP filter 209, a comparison  
5 211 is made between the output of the pseudo noise signal which represents a "desired" signal (d) and an output of the DSP filter 209 (y). A delay 213 is provided to the pseudo random noise generator so as to allow proper synchronization between noise signals as each travels by separate paths  
10 though the audio chain. As seen in FIG. 2, this chain is comprised of speaker 10, microphone 103 and DSP filter 209. An error signal (e) is produced at the output of the comparator 211 that is directed to the DSP filter 209. The error signal works to control a plurality of signal  
15 coefficients in various DSP algorithms used to process the analog signal from microphone 103. The filter coefficients are changed to provide an optimized microphone output to enable the two-way communications device to operate by having consistent gain and frequency components from the  
20 output of the its microphone 103. It will be evident to those skilled in the art that after the calibration of the microphone 103 the DSP filter 209 will continue to use the same calculated frequency coefficients in order to provide optimized audio to the communications device 101 from  
25 microphone 103. It is important to note that FIG. 2 represents a unique system identification adaptive microphone filter structure which converges directly to the inverse filter in a fixed input response (FIR) structure which has no stability issues.

30 FIG. 3 illustrates a block diagram showing the acoustic calibration of an internal speaker 301 in a portable communications device according to the preferred embodiment of the invention. FIG. 3 shows the portable communications device 101 with internal speaker 301 that is typically

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microphone calibration, the error signal works to control a plurality of signal coefficients in the DSP algorithms used to process the analog signal before entering speaker 301.

The filter coefficients are then changed to provide an  
5 optimized speaker input to enable the internal speaker 301  
in the two-way communications device to operate by having  
consistent gain and frequency components from the output of  
the its speaker 301. It will be evident to those skilled in  
the art that after the calibration of the speaker 301 the  
10 DSP filter 209 will continue to use the same calculated  
frequency coefficients in order to provide optimized audio  
to the communications device 101 from speaker 301. It is  
important to note that FIG. 4 represents a unique system  
identification adaptive speaker filter structure which  
15 converges directly to the inverse filter in a fixed input  
response (FIR) structure which has no stability issues.

While the preferred embodiments of the invention have  
been illustrated and described, it will be clear that the  
invention is not so limited. Numerous modifications,  
20 changes, variations, substitutions and equivalents will  
occur to those skilled in the art without departing from the  
spirit and scope of the present invention as defined by the  
appended claims.

What is claimed is:

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